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| 09/622,736 | 10/27/2000 | Mohammed Javed Absar | 851663.413US | 2744 |

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Please find below and/or attached an Office communication concerning this application or proceeding.

PTO-90C (Rev. 10/03)

| | Application No. | Applicant(s) |
|---|---|---|
| **Office Action Summary** | 09/622,736 | ABSAR ET AL. |
| | Examiner | Art Unit |
| | Qi Han | 2654 |

*-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --*

**Period for Reply**

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM
THE MAILING DATE OF THIS COMMUNICATION.
- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133).
Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

**Status**

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL.**     2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

**Disposition of Claims**

4)☒ Claim(s) *1-39* is/are pending in the application.

    4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1-39* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

**Application Papers**

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

    Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

    Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

**Priority under 35 U.S.C. § 119**

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

    a)☒ All   b)☐ Some * c)☐ None of:

      1.☐ Certified copies of the priority documents have been received.

      2.☐ Certified copies of the priority documents have been received in Application No. _____.

      3.☒ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

    * See the attached detailed Office action for a list of the certified copies not received.

**Attachment(s)**

1)☒ Notice of References Cited (PTO-892)

2)☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3)☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08)
Paper No(s)/Mail Date _____.

4)☐ Interview Summary (PTO-413)
Paper No(s)/Mail Date. _____ .

5)☐ Notice of Informal Patent Application (PTO-152)

6)☐ Other: _____.

## DETAILED ACTION

### *U.S.C. National Stage Application*

1.      Acknowledgement is made of the indication that the present application is filed under 35 U.S.C. 371, of the indication that the required form PCT/DO/ED/903 is present, and of the use of transmittal form PCT/DO/EO/1390. Thus, the present application is being treated as a filing under 35 U.S.C. 371.

### *Priority*

1.      Receipt is acknowledged of papers submitted under 35 U.S.C. 119(a)-(d), which papers have been placed of record in the file.

### *Specification*

2.      The disclosure is objected to because of the following informalities:

a.      on page 2, line 19, the phrase of "in a from" appears to be -- in a form-- . Appropriate correction is required.

b.      on page 10, Eq.9, an imaginary symbol "j" is missing in the middle expression of the equation. Appropriate correction is required.

c.      on page 11, lines 1 and 7, the equation of "$G_k = g_{k,r} + g_{k,i}$ "appears to be -- $G_k = g_{k,r} + j\, g_{k,i}$ -- . Appropriate correction is required.

d.      on page 11, Eq.13, the exponential factor "2k" in the left of the equation appears to be -- k -- . Appropriate correction is required.

e.      on page 14, line 3, the phrase of "in Eq. 18" appears to be -- in Eq. 20-- . Appropriate correction is required.

f.     on page 15, lines 1-2, the statement of "From the expression from Eq. 22 and 23

into Eq. 16, the MDCT for each channel is obtained" lacks sufficient mathematical proving and

reasoning because there is no $G'_k$ in the Eq. 16 and no relationship (or description) between the

y(n) and the Eq. 16.  Appropriate correction is required.

### *Claim Objections*

3.     Claim 27 is objected to because the claimed equations (Xk and Yk) lack sufficient

support in the specification.  Appropriate correction is required.

### *Rejections - 35 USC § 101*

35 U.S.C. 101 reads as follows:

> Whoever invents or discovers any new and useful process, machine, manufacture, or
> composition of matter, or any new and useful improvement thereof, may obtain a patent
> therefor, subject to the conditions and requirements of this title.

5.     Claims 1-2,4-5 and 7-8 are rejected under 35 U.S.C. 101 because the claimed invention is

directed to non-statutory subject matter.

Claim 1 defines a non-statutory method because they merely manipulate an abstract idea

(mathematical algorithm) without a claimed limitation to a practical application. The disclosed

invention has a practical application in the technological arts (audio signal coding); however, the

claimed method, a series of steps to be performed, consist solely of text of version of

mathematical operations and/or data manipulations without a claimed limitation to the practical

application and does not have any post or pre-computer process activity.

A review of the application shows the disclosed invention thereof to be an audio signal

coding method or apparatus for a multi-channel encoder.  This is a practical application within

the technological arts. However, claims 1-2,4-5 and 7-8 reviewed in light of the specification,

simply recite a mathematical algorithm for coding audio data. As can be seen by claims 1-2,4-5 and 7-8, these claims recite directly a mathematical algorithm by setting forth the steps of "multiplying the input samples with a first trigonometric function ..., computing a fast Fourier transform ..., ... , subtracting the corresponding multiplied addition ..." These steps are mathematical in nature.

Further reviewing claim 2 that depends on claim 1, it recites that the coded audio comprise DCT coefficients, which it is still based on abstract data manipulation and methodical algorithm operation, without specific application in the technological arts.

Further reviewing claims 4-5,7-8, the rejection are based on the same reason described for claim 2, because the recited limitation in claims 4-5,7-8 have same or similar nature of abstract data manipulation and methodical algorithm operation without specific application in the technological arts, as claim 2, respectively.

As per the claims, the language "A method for coding audio data" in the claim preambles, does not transform the claimed subject matter into statutory subject matter. The recital is merely a field of use or desired end use limitation.

A mathematical algorithm is not made statutory by "attempting to limit the use of the formula to a particular technological environment." Diehr, 450 U.S. at 191, 209 USPQ at 10. Thus, "field of use" or "end of use" limitations in the claim preamble are insufficient to constitute a statutory process.

The above review of the claims shows that the subject matter claimed in addition to the mathematical algorithm is not sufficient on its own to render the claims as a whole statutory.

Further, a review of each claim as a whole fails to show the transformation or reduction of subject matter to a different state or thing.

It is readily apparent that when claims 1-2, 4-5 and 7-8 are each taken as a whole, the claims are directed to the preemption of a mathematical algorithm, and thus are non-statutory.

6.     To expedite a complete examination of the instant application the claims rejection under

35 U.S.C 101 (nonstatutory) above are further rejected as set forth below in anticipation of

applicant amending these claims to place them within the four statutory categories of invention.

### *Claim Rejections - 35 USC § 103*

The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all

obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

4.     Claims 1-9 and 17-23 are rejected under 35 U.S.C. 103(a) as being unpatentable over

Fielder et al. (US 5,479,562) hereinafter referenced as Fielder.

As per **claim 1**, Fielder discloses a method and apparatus for encoding and decoding

audio information, comprising:

i)      multiplying the input samples with a first trigonometric function factor to

generate an intermediate sample sequence (column 35, line 35 to column 36, line 8 and equation

(26), 'premultiply step') ;

ii)     computing a fast Fourier transform of the intermediate sample sequence to

generate a Fourier transform coefficient sequence (column 36, lines 9-19 and equation (27) );

iii)     for each transform coefficient in the sequence, multiplying the real and imaginary

components of the transform coefficient by respective second trigonometric function factors,

adding the multiplied real and imaginary transform coefficient components to generate an

addition stream coefficient, and subtracting the multiplied real and imaginary transform

coefficient components to generate a subtraction stream coefficients (column 36, lines 20-35 and

equation (28), 'a postmultiply step', wherein equation (28) has equivalent function and same

result as the equation 16 in the specification which reads on the claimed limitation);

iv) multiplying the addition and subtraction stream coefficients with respective third

trigonometric function factors (column 36, lines 20-35 and equation (28), 'a postmultiply step',

wherein equation (28) has equivalent function and same result as the equation 16 in the

specification which reads on the claimed limitation); and

v) subtracting the corresponding multiplied addition and subtraction stream coefficients

to generate audio coded frequency domain coefficients (column 36, lines 20-35 and equation

(28), 'a postmultiply step', wherein equation (28) has equivalent function and same result as the

equation 16 in the specification which reads on the claimed limitation);

But, Fielder does not expressly disclose all intermediate results or steps in processing.

However, those intermediate results or steps of the processing are obvious to one skilled in the

art to be obtained through a simple mathematical reasoning, because Fielder teaches a starting

equitation (24), condition equations (6 and 25), intermediate equations (26, 27 and some part of

28), and final result equation (28), which corresponds to Eq.1, Eq. 13 and Eq. 16 of the

specification and can derive or inherently include other intermediate steps of using trigonometric

functions, multiplying addition and subtraction processes as claimed. Therefore, it would have

been obvious to one of ordinary skill in the art at time the invention was made to compute

Fielder' the equations by a simple mathematical reasoning, for the purpose of providing a

complete computation algorithm to reach the final conclusion (equation 28) from a staring point

(equation 24) (Fielder: column 35, lines 60-67).

As per **claim 2** (depending on claim 1), Fielder further discloses the audio coded

frequency domain coefficients comprise modified discrete cosine transform coefficients (column

35, lines 33-59).

As per **claim 3** (depending on claim 1), Fielder further discloses that the first

trigonometric function factor for each audio sample is a function of the audio sample sequence

position (n) and the number (N) of samples in the sequence (column 36, eq. (26)).

As per **claim 4** (depending on claim 1), Fielder further discloses that the respective

second trigonometric function factors for each transform coefficient in the sequence are

respective functions of the transform coefficient sequence position (k) and the number (N) of

coefficients in the sequence (column 36, eq. (28)).

As per **claim 5** (depending on claim 1), Fielder further discloses that the respective third

trigonometric function factors are respective functions of the transform coefficient sequence

position (k) (column 36, eq. (28) and column 18 eq. (6), angle of 2pi (k+ ½) m/N = 2pi (k+ ½)/4

+ pi (k+ ½), where m=(N/2 +1)/2, so that a trigonometric function factor of sum of two angles

can be expressed by a sum (two terms) of trigonometric function factors with individual angles,

i.e. cos(a+b) = cos (a) cos(b) + sin(a)sin(b), therefore one of the trigonometric function factors

reads on the claimed limitation).

As per **claim 6** (depending on claim 1), Fielder does not expressly disclose that the step i)

comprises multiplying the input sequence samples x[n] by the first trigonometric function factor

cos(πn/N) to generate the intermediate sample sequence, where: x[n] are the input sequence

audio samples; N is the number of input sequence audio samples. However, Fielder discloses

multiplying the input sequence samples x[n] by cos(- $\pi n/N$) (column 36, eqs. (26) and (27),

where exp(-j $\pi n/N$) = cos(- $\pi n/N$) + j sin (- $\pi n/N$) ), which is sympatric to eq. 11 in the

specification (page 10) and also corresponds to eq. 13 (specification: page 11), which suggests

that computing intermediate result in eq. 13 by using factor exp(-j $\pi n/N$) has equivalent

functionality of computing intermediate result in eq. 12, since there is a conjugation relationship

between them). Therefore, it would have been obvious to one of ordinary skill in the art at time

the invention was made to modify Fielder by providing multiplying the input sequence samples

x[n] by an exchangeable a factor (either exp(j $\pi n/N$) or exp(j $\pi n/N$), for the purpose of providing

a alternative computation for the process.

As per **claim 7** (depending on claim 1), Fielder further discloses that step ii) comprises

computing the fast Fourier transform of the intermediate sample sequence so as to generate said

transform coefficient sequence Gk = gk,r + jgk,i, where: Gk is the transform coefficient

sequence; gk,r are the real transform coefficient components; gk,i are the imaginary transform

coefficient components; and k=O  (N/2 - 1), (column 36, eqs. (27) and (28), where X*(k), R(k)

and Q(k) correspond to Gk, gk,r and gk,i, respectively).

As per **claim 8** (depending on claim 1), Fielder does not expressly disclose that step iii)

comprises determining the addition stream coefficients T2 and subtraction stream coefficients T,

according to: T1 = gk,r cos (pi(k + 1/2)/N) - gk,i sin (pi(k + 1/2)/N); T2 =gk,r cos(pi(k+1/2)/N)

+ gk,i sin (pi(k+1/2)/N); where T1 and T2 are the subtraction stream and addition stream

coefficients, respectively. However, Fielder discloses, as stated above (see claim 1), a starting

equitation (24) with condition equations (6 and 25) that corresponds to eq. 1 of the specification,

intermediate equations (26, 27) that corresponds to intermediate eq. 13 of the specification, and

final result equation (28) that corresponds to eq. 16 of the specification, in which the claimed

limitation can be proven or derived from the referenced prior art equations through a

mathematical reasoning, which also means that the equation 28 can be reasoned to derive the

claimed terms. Further, according to eq. (28) (column 36) and eq. (6) (column 18), the angle in

eq. (28) can be separated into to two angles: 2pi (k+ ½) m/N = 2pi (k+ ½)/4 + pi (k+ ½), where

m=(N/2 +1)/2, so that the trigonometric function factor of sum of two angles can be expressed by

a sum (two term) of trigonometric function factors of individual angles, such as cos(a+b) = cos

(a) cos(b) + sin(a)sin(b), thus one of the trigonometric function factors can read on the claimed

limitation. Therefore, it would have been obvious to one of ordinary skill in the art at time the

invention was made to compute Fielder' the equations by complete a mathematical reasoning,

including intermediate steps and result as claimed, for the purpose of providing a complete

computation algorithm from a staring point (equation 24) to reach the final conclusion (equation

28) (Fielder: column 35, lines 60-67).

As per **claim 9** (depending on claim 1), Fielder does not expressly disclose that steps iv)

and v) comprise generating the audio coded frequency domain coefficients Xk according to: Xk

= T1 cos (pi(2k+1)/4) - T2 sin (pi(2k+1)/4); where Xk are the audio coded frequency domain

coefficients; and cos (pi(2k+1)/4) and sin (pi(2k+1)/4) are the third trigonometric function

factors. However, Fielder discloses, as stated above (see claim 1), a starting equitation (24) with

condition equations (6 and 25) that corresponds to eq.1 of the specification, intermediate

equations (26, 27) that corresponds to intermediate eq. 13 of the specification, and final result

equation (28) (see column 35, line 33 to column 36, line 35) that corresponds to eq. 16 of the

specification, in which the claimed limitation can be proven or derived from the referenced prior

art equations through a mathematical reasoning, which suggests that the equation can be

reasoned to derive the claimed terms. Further, according to eq. (28) (column 36) and eq. (6)

(column 18), the angle in the eq. (28) can be separated into to two angles: 2pi (k+ ½) m/N = 2pi

(k+ ½)/4 + pi (k+ ½), where m=(N/2 +1)/2, so that the trigonometric function factor with a sum

of two angles can be expressed by a sum (two term) of trigonometric function factors with

individual angles, such as cos(a+b) = cos (a) cos(b) + sin(a)sin(b), thus, one of the trigonometric

function factors can read on the claimed limitation. Therefore, it would have been obvious to

one of ordinary skill in the art at time the invention was made to compute Fielder' the equations

by a mathematical reasoning, including intermediate steps and result as claimed, for the purpose

of providing a complete computation algorithm from a staring point (equation 24) to reach the

final conclusion (equation 28) (Fielder: column 35, lines 60-67).

As per **claim 17**, the rejection is based on the same reason described for claim 1, because

claim 17 recites same or similar limitation(s) as claim 1.

As per **claim 18** (depending on claim 17), Fielder further discloses that the pre-

multiplication factor, and first and second post-multiplication factors are trigonometric function

factors (column 36, equations (26) and (28), wherein factor exp(-j πn/N) = cos(- πn/N) + j sin (-

πn/N), and term of cos [2pi (k+ ½) m/N] = cos [2pi (k+ ½)/4 + pi (k+ ½)] when using equation

16: m=(N/2 +1)/2; so that by further using a trigonometric property of cos(a+b) = cos (a) cos(b)

+ sin(a)sin(b), the term of cos() (or sin()) in equation 28 produces two trigonometric function

factors, which reads on the claimed limitation).

As per **claims 19-21** (depending on claim 17), the rejection is based on the same reason

described for claims 3-5 respectively, because claims 19-21 recites same or similar limitation(s)

as claims 3-5 respectively.

As per **claim 22** (depending on claim 17), Fielder further discloses that the pre-

processing operations are performed on each sample in the input sequence individually (column

36, equation 27, which shows that the operation is performed on each sample in input $x(n)$

individually).

As per **claim 23** (depending on claim 17), Fielder further discloses that the post-

processing operations are performed on each transform coefficient in the sequence individually,

(column 36, equation 28, which shows that the post-processing operation is performed on each

transform coefficient $R(k)$ and $Q(k)$ individually).


5.      Claims 10-13, 16 and 24-27 are rejected under 35 U.S.C. 103(a) as being unpatentable

over Fielder in view of Proakis et al. ("Digital Signal Processing, principles, algorithms, and

applications", 3rd Edition, 1996, ISBN 0-13-373762-4) hereinafter referenced as Proakis.

As per **claim 10**, Fielder discloses a method and apparatus for encoding and decoding

audio information, comprising:

combining first and second sequences of digital audio samples from first and second

audio channels into a single complex sample sequence (column 16, line 40 to column 17, line 11

'a single FFT can be used to perform the DCT and DST simultaneously by define them

respectively as the real and imaginary components of a signal complex transform' and

'processing a signal sample block from each of the two channels', which suggest that the signal

uses the real component for one channel and imaginary components for another channel);

determining a Fourier transform coefficient sequence (column 16, lines 40-55, 'a single

FFT can be used to perform the DCT and DST simultaneously by define them respectively as the

real and imaginary components of a signal complex transform'; column 36, lines 9-35 and

equations 27 and 28, where in the signal x(n) has real and imaginary components:

$x(n)=xr(n)+jxi(n))$;

generating first and second transform coefficient sequences by combining and/or

differencing first and second selected transform coefficients from said Fourier transform

coefficient sequence (column 16, lines 40-55, 'a single FFT can be used to perform the DCT and

DST simultaneously by define them respectively as the real and imaginary components of a

signal complex transform'; column 36, lines 9-35 and equations 27 and 28, 'signal sample blocks

from the two channels are transformed by FFT processes into DCT1/DCT2 block pair', wherein

equation 27 includes two components: $X^*(k)= FFT[xr(n)exp(-jnpi/N)+jxi(n)exp(-jnpi/N)] =$

$R(k)+jQ(k)$ ); and

for each of the first and second transform coefficient sequences, generating audio coded

frequency domain coefficients to generate respective sequences of said audio coded frequency

domain coefficients for the first and second audio channels (column 16, lines 40-55, 'a single

FFT can be used to perform the DCT and DST simultaneously by define them respectively as the

real and imaginary components of a signal complex transform'; column 36, lines 20-55 and

equation 28, 'In two-channel systems, signal sample blocks from each of two channels are

transformed by FFT processes into DCT1/DCT2 block pair').

Even though, Fielder discloses that a single FFT can be used in to perform the DCT and

DST simultaneously by defining them respectively as the real and imaginary components of a

single complex transform, as stated above, Fielder does not expressly disclose the detailed

intermediate steps for computing and/or reasoning the process of a two-channel system using

complex signal in an FFT transform. However, this feature is well known in the art as evidenced

by Proakis, who teaches symmetry properties of the discrete-time Fourier transform (page 290-

291) that disclose the mathematical relationships between different time domain/frequency

domain signal components, and efficient computation of the DFT of two real sequences (page

475-476) that can compute two real signal sequences in a complex-valued sequence by

performing a single DFT (FFT), so that the respective sequences of audio frequency domain

coefficient sequences for the two real signal sequences (corresponding to two audio channel

signals) can be derived by using the FFT transformed coefficients and the symmetry properties.

Therefore, it would have been obvious to one of ordinary skill in the art at time the invention was

made to modify Fielder by specifically providing a FFT algorithm to perform a single DFT for

two real signal (two channel) sequences by using the symmetry properties of the Fourier

transform, as taught by Proakis, for the purpose of enhancing the efficiency of the FFT algorithm

(Proakis: page 475, paragraph 6).

As per **claim 11** (depending on claim 10), Fielder in view of Proakis further discloses that

for each corresponding coefficient in the first and second transform coefficient sequences,

selecting first and second transform coefficients from said Fourier transform coefficient

sequence, determining a complex conjugate of said second transform coefficient, combining said

first transform coefficient and said complex conjugate for said first transform coefficient

sequence and differencing said first transform coefficient and said complex conjugate for said

second transform coefficient sequence, (Fielder: column 36, lines 35 and equations 27, 28 and 6;

Proakis: pages 290-291, equation 4.3.37 and Table 4.4, wherein two time domain signal

sequences can be defined as a complex sequence: $x(n) = xr(n) + jxi(n)$; the frequency domain

sequence can be expressed by: $X(k)=FFT[xr(n)exp(-jnpi/N)+jxi(n)exp(-jnpi/N)] = Xr(k)+jXi(k)$,

which corresponds to equation 27 of Fielder; and the frequency domain sequence can be further

expressed by: $X(k)= Xr(k)+jXi(k) = [Xre(k)+jXio(k)] + [Xro(k)+jXie(k)] = Xxr(k) +Xxi(k)$,

wherein the subscripts indicate: r -- real part, i –imaginary part, e – even part, o – odd part, x –

corresponding input signal sequences, which corresponds to term $R(k)$ and $Q(k)$ in equation 28

of Fielder by combining symmetry properties on equation 4.4.37 (Proakis: page 290) and

complex conjugate process in Table 5.1 (Proakis: page 415)).

As per **claim 12** (depending on claim 10), the rejection is based on the same reason

described for claim 6, because claim 12 recites same or similar limitation(s) as claim 6.

As per **claim 13** (depending on claim 11), as best understood in view of disclosure

objection f. (see above), Fielder in view of Proakis further discloses a properties of DTFT:

$Xe(k)=1/2[X(k)+X^*(N-k)]$ and $Xo(k)=1/2[X(k)-X^*(N-k)]$ (Proakis: page 415, Table 5.1) and the

derived equations for computation of the DFT of two real sequences (Proakis: page 476,

equations 6.2.7 and 6.2.8), where e indicates even part, o indicates odd part, and $X(k)$

corresponds to coefficient $X^*(k)$ in equation 27 of Fielder (Fielder: column 36, lines 1-35), so

that the combined teachings correspond to the clamed "said first and second transform

coefficient sequences are generated according to: Gk (Zk + Z*N-k-1)/2, G'k (Zk - Z*N-k-1)/2j

where Gk is said first transform coefficient sequence; G'k is said second transform coefficient

sequence; N is the number of input sequence audio samples; $k = 0,\ldots, (N/2 - 1)$; Zk is said first

transform coefficient; $Z*N-k-1$ is the complex conjugate of said second transform coefficient;

and j is the complex constant".

As per **claim 16** (depending on claim 10), Fielder in view of Proakis further discloses

applying a windowing function in combination with multiplying the complex sample sequence

by a first trigonometric function factor (Fig. 1a, 'analysis widow'; Figs. 6a-6d).

As per **claim 24**, it recites audio coding method, which corresponds to the combination of

claims 1, 10 and 13. The rejection is based on the same reason described for claims 1,10 and 13,

because claim 24 recites same or similar limitation(s) as claims 1,10 and 13.

As per **claims 25** (depending on claim 24), the rejection is based on the same reason

described for claim 3, because claim 25 recites same or similar limitation(s) as claim 3.

As per **claims 26** (depending on claim 24), the rejection is based on the same reason

described for claim 18, because claim 26 recites same or similar limitation(s) as claim 18.

As per **claim 27**, as best understood in view of claim objection (see about), Fielder

discloses a method and apparatus for encoding and decoding audio information, comprising:

obtaining first and second input sequences of digital audio samples x[n], y[n]

corresponding to respective first and second audio channels, (column 16, line 40 to column 17,

line 11, 'both input signal sample blocks consist only real-valued samples' and 'processing a

signal sample block from each of the two channels');

combining the first and second input sequences of digital audio samples into a single

complex input sample sequence z[n], where $z[n] = x[n] + jy[n]$, (column 16, lines 43-64; 'a

single FFT can be used to perform the DCT and DST simultaneously by define them respectively

as the real and imaginary components of a signal complex transform', which suggest that the

input signals includes a real component for one channel and an imaginary components for

another channel);

pre-processing the complex input sequence samples including applying a pre-

multiplication factor cos(pi n/N) +jsin(pi n/N) to obtain modified complex input sequence

samples, where N is the number of audio samples in each of the first and second input sequences

and n = 0,..., (N-1) (column 36, equations (26) and (27), wherein a factor of exp(-j $\pi$n/N) =

cos(- $\pi$n/N) + j sin (- $\pi$n/N) is used for pre-multiplying, which has equivalent functionality of the

claimed limitation because one of intermediate results can be derived from the other by

conjugate processing (based on eqs. 12 and 13 of the specification) );

transforming the modified complex input sequence samples into a complex transform

coefficient sequence Zk utilizing a fast Fourier transform, wherein k = 0,..., (N/2- 1), (column

16, lines 40-55, 'a single FFT can be used to perform the DCT and DST simultaneously by

define them respectively as the real and imaginary components of a signal complex transform';

column 36, lines 9-35 and equations 27, 'signal sample blocks from the two channels are

transformed by FFT processes into DCT1/DCT2 block pair', wherein input sequence x(n) may

have two sequences combined in a complex sequence, so that in this case it is obvious that

equation 27 would be: X*(k)= FFT[xr(n)exp(-jnpi/N) + jxi(n)exp(-jnpi/N)]; and

post-processing the sequence of complex transform coefficients to obtain first and second

sequences of audio coded frequency domain coefficients (column 36, lines 9-35 and equations

28).

But, Fielder does not expressly disclose "to obtain first and second sequences of audio

coded frequency domain coefficients corresponding to the first and second audio channels Xk,

Yk" according to claimed intermediate steps of computing the process for a two-channel system

using complex signal in an FFT transform. However, this feature is well known in the art as

evidenced by Proakis, who teaches symmetry properties of the discrete-time Fourier transform

(page 290-291) that disclose the mathematical relationships between different time

domain/frequency domain signal components, and efficient computation of the DFT of two real

sequences (page 475-476) that combines the two real signal sequences, such as two channel

signal sequences, into a complex-valued sequence for performing a single DFT (FFT), so that the

respective sequences of audio frequency domain coefficient sequences for the two real signal

sequences (corresponding to two audio channel signals) can be derived by using the FFT

transformed coefficients and the symmetry properties. Particularly, Proakis discloses equations

6.2.7 and 6.2.8 (page 476) that are equivalent to the claimed Gk and G'k, and symmetry equation

5.2.31 (page 415), which can be used in equation 28 of Fielder to generate the claimed result by

mathematically reasoning: let $a=2pi(k+1/2)/4$, $b=pi(k+1/2)/N$, $Xr=R(k)$, $Xi= -Q(k)$,

$X(k)=(X*(k))*=[R(k)-jQ(k)]= Xr+jXi$ (Fielder: eq. 27); and take $m=(1+N/2)/2$ (Fielder: eq.16),

then from eq.28 (Fielder):

$$C(k)= R(k)\cos (a+b) + Q(k)\sin(a+b) = Xr \cos (a+b) - Xi \sin(a+b)$$

$$= Xr [\cos (a)\cos (b) - (\sin(a)\sin(b)] - Xi[\sin(a)\cos(b)+\cos(a)\sin(b)]$$

$$=\cos(b)[Xr \cos(a) - Xi \sin(a)] - \sin(b)[Xr \sin (a) + Xi \cos(a)]$$

$$= \cos(b)[(Xre+Xro)\cos(a)-(Xie+Xio)\sin(a)] - \sin(b)[(Xre+Xro)\sin(a)+(Xie+Xio)\cos(a)]$$

where, subscripts indicate: r—real part, i—imaginary part, e—even part, o—odd part, since

Proakis teaches that even part of frequency coefficient: $X1(k) = [X(k)+(X*(N-k)]/2=Xre+jXie$

corresponds to real part of input sequence $x1(n)$ and odd part of frequency coefficient

$X2(k)=[X(k)+(X*(N-k)]/j2=Xro+jXio$ corresponds to imaginary part of input sequence $x2(n)$

(Proakis: page 415, Table 5.1 and equation 5.2.31; page 476, equations 6.2.7 and 6.2.8), thus,

separating the even and odd parts into two groups would lead to obtain the respective frequency

coefficients corresponding to the two input signal sequences as claimed.

Therefore, it would have been obvious to one of ordinary skill in the art at time the

invention was made to modify Fielder by specifically providing a FFT algorithm by performing a

single DFT for two real signal (two channel) sequences by using the symmetry properties of the

Fourier transform, as taught by Proakis, for the purpose of enhancing the efficiency of the FFT

algorithm (Proakis: page 475, paragraph 6).


6.      Claims 14-15 and 28-39 are rejected under 35 U.S.C. 103(a) as being unpatentable over

Fielder in view of Proakis and further in view of Jhung (US 6304847 B1).

As per **claims 14** (depending on claim 10), even though Fielder teaches the tradeoff of

using longer or shorter block length for a transform (column 3, lines 30-67), Fielder in view of

Proakis does not expressly disclose "examining said first and second sequences of digital audio

samples to determine a short or long transform length, and coding the audio samples using a

short or long transform length as determined". However, this feature is well known in the art as

evidenced by Jhung, who discloses that the Dolly AC-3 standard utilizes long transform or two

short transform based on the transition condition (column 3, line 62 to column 4, line 24).

Therefore, it would have been obvious to one of ordinary skill in the art at time the invention was made to modify Fielder in view of Proakis by specifically providing long transform or two short transform based on the transition condition, as taught by Jhung, for the purpose of handling different transition situations (Proakis: column 3, line 63 to column 4, line 2).

As per **claims 15** (depending on claim 10), the rejection is based on the same reason described for claim 14, because claim 15 recites same or similar limitation(s) as claim 14.

As per **claims 28-39**, they recite an apparatus for coding input audio samples. The rejection is based on the same reason described for claims 1-2, 18, 3-5, 22-23, 14, 10 and 38-39 respectively, because claims 28-39 recite same or similar limitation(s) as claims 1-2, 18, 3-5, 22-23, 14, 10 and 38-39 respectively.

## *Conclusion*

7.     Any response to this office action should be mailed to:
Commissioner of Patents and Trademarks, P.O. Box 1450, Alexandria, VA22313-1450
or faxed to:
    (703)-872-9314
Hand-delivered responses should be brought to:
    Crystal Park II, 2121 Crystal Drive, Arlington. VA. Sixth Floor (Receptionist).

Any inquiry concerning this communication or earlier communications from the examiner should be directed to I Han whose telephone numbers is (703) 305-5631. The examiner can normally be reached on Monday through Thursday from 9:00 a.m. to 7: p.m.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Richmond Devil, can be reached on (703) 305-6954.

Any inquiry of a general nature of relating to the status of this application or proceeding should be directed to the Technology Center 2600 Customer Service Office whose telephone number is (703) 306-0377.

QH/qi
April 15, 2004

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SUPERVISORY PATENT EXAM